



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.2 and Acme Packet Net-Net 6.2.0 with AT&T IP Transfer Connect SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R5.2.1 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager R6.2, and the Acme Packet Net-Net 3800 with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.2 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.2 and the AT&T IP Transfer Connect service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect calls to alternate destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. In addition, the Avaya Aura® Communication Manager NCR and SIP User-to-User Information (UUI) features are utilized, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UUI within SIP signaling messages to the alternate destinations.

Note that these Application Notes are intended to supplement the separate document: *Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0*

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager SIP Network Call Redirection (NCR), Avaya Aura® Session Manager, and the Acme Packet Net-Net 3800¹ Session Border Controller (SBC) with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.2 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager R5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.2 and the AT&T IP Transfer Connect service and is used not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect calls to alternate² destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. The AT&T IP Transfer Connect service is typically used by enterprises that have multiple call centers that are separated geographically or otherwise not interconnected. Using SIP NCR, trunk-to-trunk routing of certain inbound calls at Avaya Aura® Communication Manager can be avoided by requesting that the AT&T network transfer the inbound caller to an alternate destination. Both intra-site and IP Transfer Connect call scenarios were tested.

Note: The AT&T IP Transfer Connect service does not support rerouting of inbound calls to international destinations. Please contact AT&T for service availability in your area.

In addition, the Avaya Aura® Communication Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI within SIP signaling messages to the alternate destinations. This capability is used in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service to transmit a limited amount of call-related data between call centers to support enhanced, customer-friendly applications and/or support efficient use of call center resources. Examples of UUI data might include a customer account number obtained during a database query and the best service routing data exchanged between Avaya Aura® Communication Manager systems.

Note that these Application Notes are intended to supplement the separate document:
Application Notes for Avaya Aura® Communication Manager R5.2.1, Avaya Aura® Session Manager R6.2 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0.

¹ Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

² Note that this is NOT the same as the “Alternate Destination Routing (ADR)” service option available with the AT&T IP Transfer Connect service.

2. General Test Approach and Test Results

The test environment consisted of:

1. A simulated enterprise with System Manager, Session Manager, Communication Manager, Avaya phones, fax machines (Ventafax application), Acme Packet Session Border Controller (SBC), and Avaya Modular Messaging.
2. A laboratory version of the AT&T IP Transfer Free service network with Transfer Connect option, to which the simulated enterprise was connected via **AVPN** or **MIS-PNT** transport.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for sample call flows) between Session Manager, Communication Manager, Acme Packet SBC and the AT&T IP Transfer Connect service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network. The following features were tested as part of this effort:

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs/Vectors, agents and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Inbound AT&T IP Transfer Connect service calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to the AT&T IP Transfer Connect service for redirection to alternate destination.
- Redirected AT&T IP Transfer Connect service calls per above arriving on Communication Manager VDNs, agents, and phones, i.e., Communication Manager as the target party for the redirected calls.
- Recovery from unsuccessful post-answer redirection attempts per above due to busy or error conditions on the alternate destination.
- Call and two-way talkpath establishment between callers and Communication Manager agents/phones.

2.2. Test Results and Known Limitations

1. Communication Manager 5.2.1 issues a BYE upon receipt of a Notify/180 during Refer calls. While this did not prevent the Refers from completing successfully, this is not desired behavior (desired behavior is for Communication Manager to wait for the AT&T IP Transfer Connect service to issue a Notify/200OK before sending the BYE, or wait until the AT&T IP Transfer Connect sends the BYE). Avaya is investigating the issue.
2. The Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (**Section 6.2**). With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service. Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services, via the same Communication Manager environment, must use the procedures described in **Section 7** of this document to alleviate potential issues with the AT&T IP Flexible Reach service. It describes having the Acme Packet SBC replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.
3. Vectors in Communication Manager can specify a maximum of 96 characters for UUI.

The test objectives stated in **Section 2.1** with limitations as noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Transfer Connect service by calling (888) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

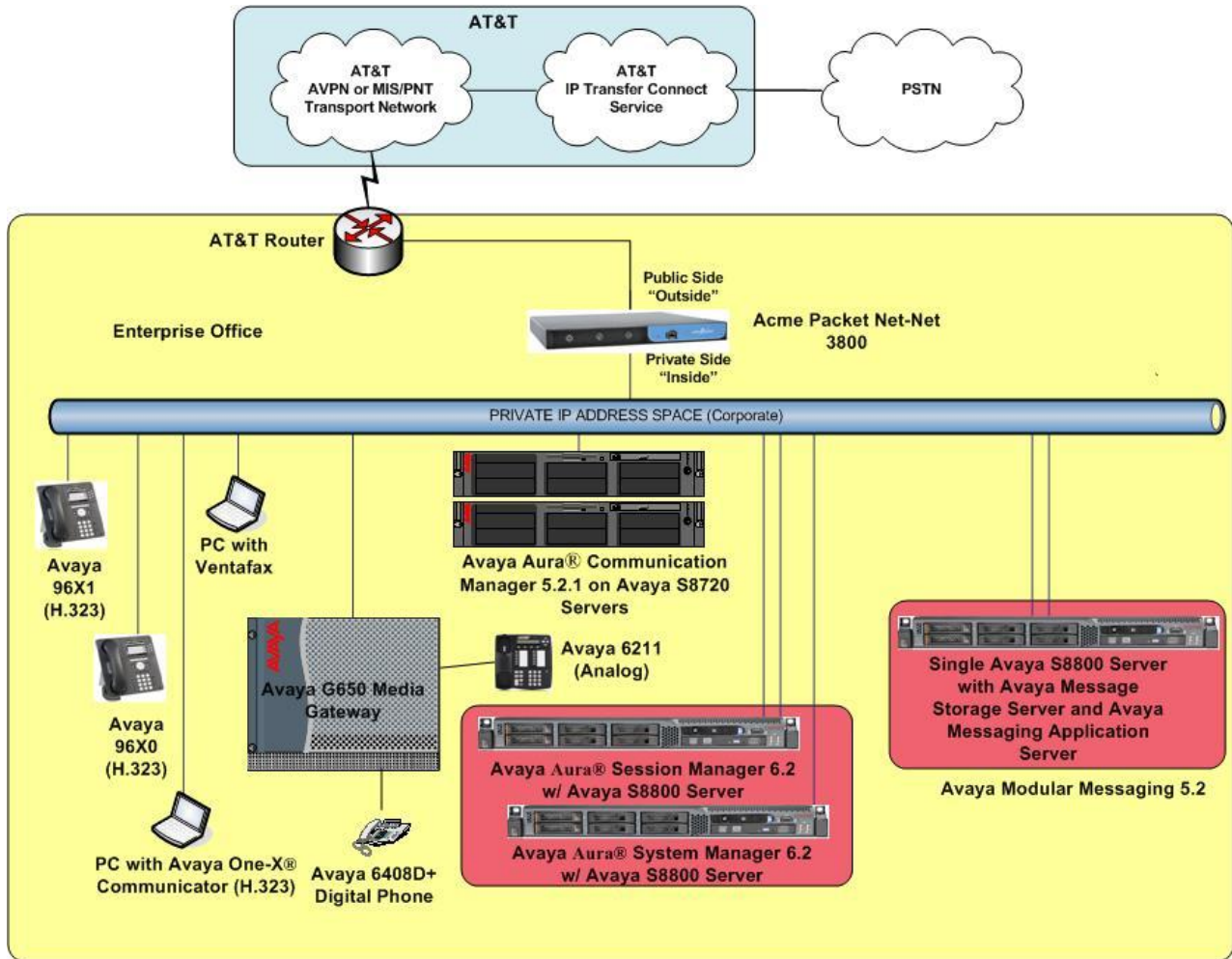
3. Reference Configuration

The reference configuration used in these Application Notes is shown in figure below and consists of several components:

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communication services for a particular enterprise site. In the reference configuration, Communication Manager 5.2.1 runs on an Avaya S8720 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. The G650 contains system boards such as the Control LAN (C-LAN) and Media Processor (MedPro). This solution is extensible to other Avaya Media Gateways.
- Avaya “desk” telephones are represented with Avaya 96x0 and 96x1 Series IP Telephones running H.323, Avaya 6408D Series Digital Telephone, Avaya Analog phone and Avaya one-X® Agent, a PC based softphone.
- The Acme Packet SBC 3800³ provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Transfer Connect service and the enterprise internal network⁴. UDP transport protocol is used between the Acme Packet SBC and the AT&T Transfer Connect service.
- An existing Avaya Modular Messaging system provides the corporate voice messaging capabilities in the reference configuration. **The provisioning of Modular Messaging is beyond the scope of this document and is shown here for illustrative purposes only.**
- Inbound calls were placed from PSTN to the Customer Premises Equipment (CPE) via the AT&T IP Transfer Connect service, through the Acme Packet SBC, Session Manager, and Communication Manager. Communication Manager terminated the calls using appropriate phone or fax stations. The H.323 phones in the CPE registered to the Communication Manager C-LANs.

³ Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

⁴ The AT&T IP Transfer Connect service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Acme Packet SBC in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme Packet SBC and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.



Reference configuration

3.1. Illustrative Configuration Information

The specific values used in this reference configuration listed in **Table 1** below are **for illustrative purposes only**. Customers must obtain and use the specific values for their configurations. For security purposes, real IP addresses and DNIS are not included.

Note - The AT&T IP Transfer Connect service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Transfer Connect service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® System Manager	
Management IP Address	10.80.150.209
Avaya Aura® Session Manager	
Management IP Address	10.80.150.210
Network IP Address	10.64.19.210
Avaya Aura® Communication Manager	
Control LAN (C-LAN) IP Address	10.80.130.206
Media Processor (MedPro) IP Address	10.80.130.207
Skill (Hunt Group) Extensions	53xxx
Agent Extensions	53xxx
Phone Extensions	50xxx
Announcement Extensions	33xxx
Vector Directory Numbers (VDN)	20xx
Acme Packet Session Border Controller	
IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Transfer Connect service)	192.168.62.50
IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager)	10.80.130.250
AT&T IP Transfer Connect service	
Border Element IP Address	192.242.225.210
Digits Passed in SIP Request-URI (RRN)	000004154891089 (SIP Redirect) 000004154901090 (SIP Refer Attended) 000004154911091 (SIP Refer Unattended) 000004154921092, 000004154931093 (Speed Dial Codes used for Redirection/Refer)

Table 1: Illustrative Values Used in this Reference Configuration

3.2. Call Flows

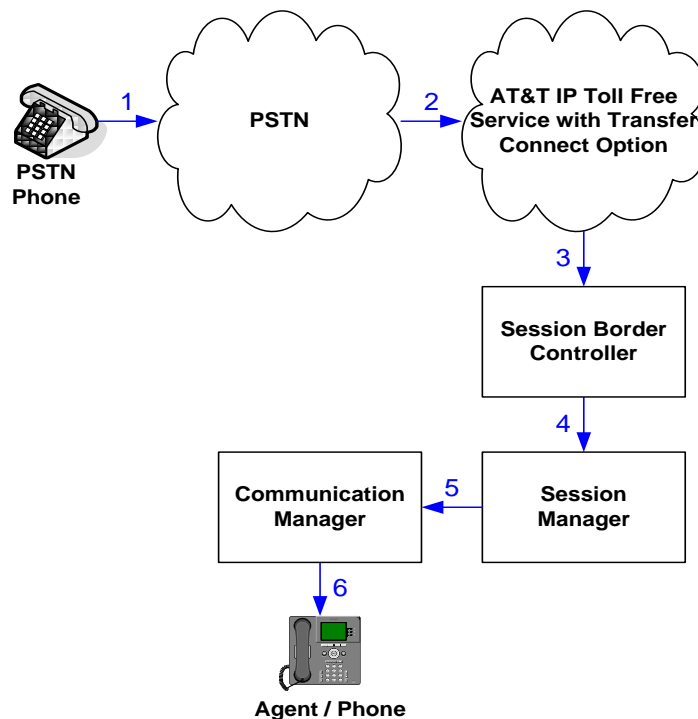
To understand how AT&T IP Transfer Connect service calls are handled by Session Manager and Communication Manager, four basic call flows are described in this section, however for brevity not all possible call flows are described.

Note: In the call scenarios that follow, the term “alternate destination” does NOT refer to the “Alternate Destination Routing (ADR)” service option of the AT&T IP Toll Free service. ADR and the AT&T IP Transfer Connect service are unrelated.

3.2.1. Inbound Call to an Agent/Station using VDN/Vectors

The first call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a phone, fax, or in some cases, a vector.

1. A PSTN phone originates a call to an AT&T IP Transfer Connect service number.
2. The PSTN routes the call to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect service routes the call to Session Border Controller.
4. The Session Border Controller performs any necessary SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to
 - A vector, which in turn, routes the call to an agent
 - Directly to an agent or a phone/fax extension.

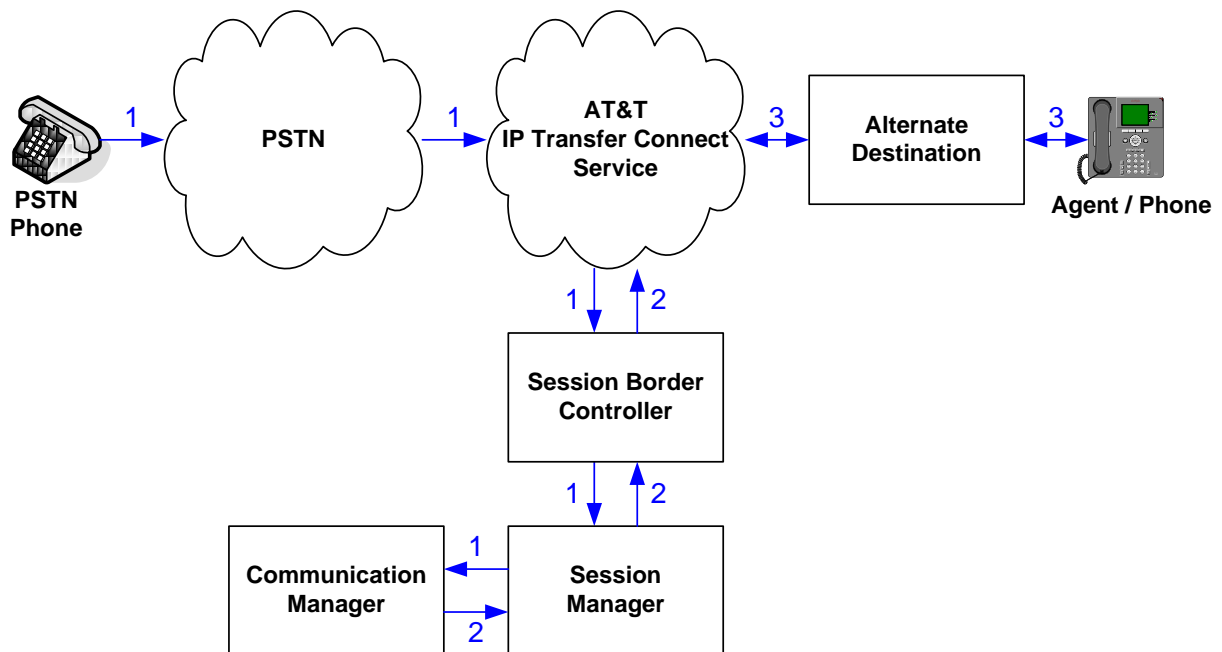


Inbound AT&T IP Transfer Connect Call to VDN/Agent/Phone

3.2.2. Inbound Call Redirection to Alternate Destination

The second call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector, without answering the call, immediately redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

1. Same as the first five steps from the first call scenario.
2. Communication Manager routes the call to a vector, which redirects the call by sending a SIP 302 message back out on the SIP trunk on which the inbound call arrived. The SIP 302 message is routed back through Session Manager and then the SBC to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (Communication Manager) is no longer involved in the call whether the redirection succeeds or fails, and thereby releases the trunk.
3. The AT&T IP Transfer Connect service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).

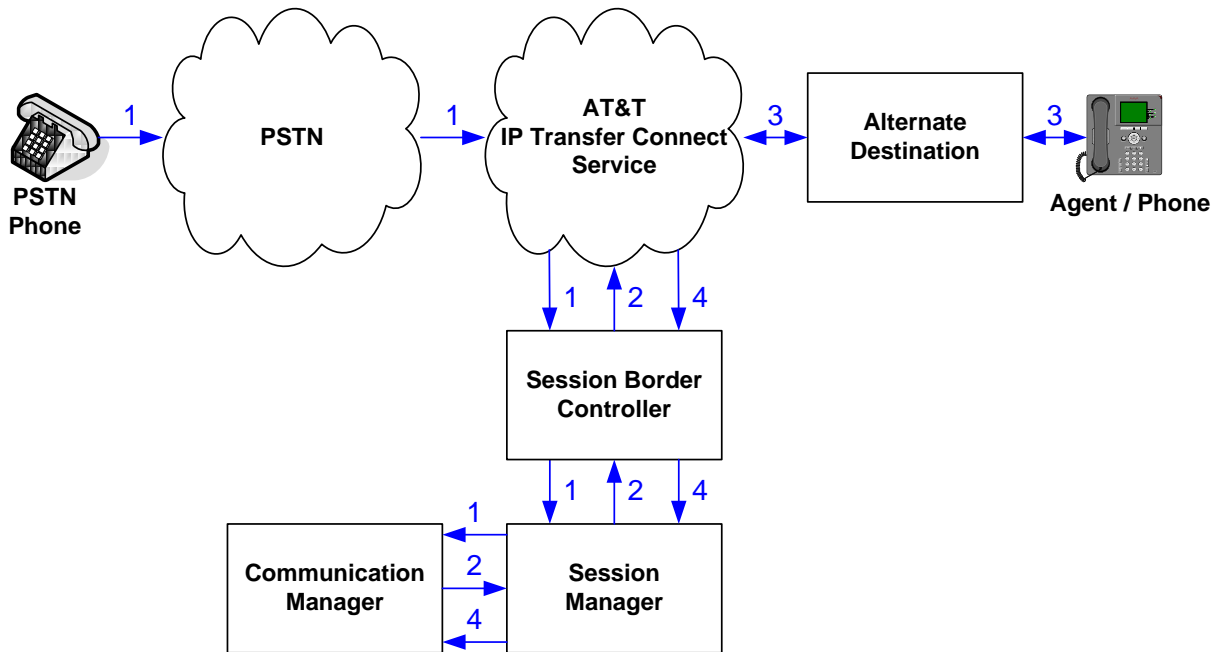


Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection Messaging

3.2.3. Inbound Call Referred to Alternate Destination

The third call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

1. Same as the first five steps from the first call scenario.
2. Communication Manager routes the call to a vector, which answers the call and attempts to redirect the call by sending a SIP REFER message back out on the SIP trunk on which the inbound call arrived. The SIP REFER message specifies the alternate destination, and is routed back through Session Manager and then the SBC to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination) and upon answer, connects the calling party to the target party.
4. The AT&T IP Transfer Connect service clears the call on the redirecting/referring party (Communication Manager).



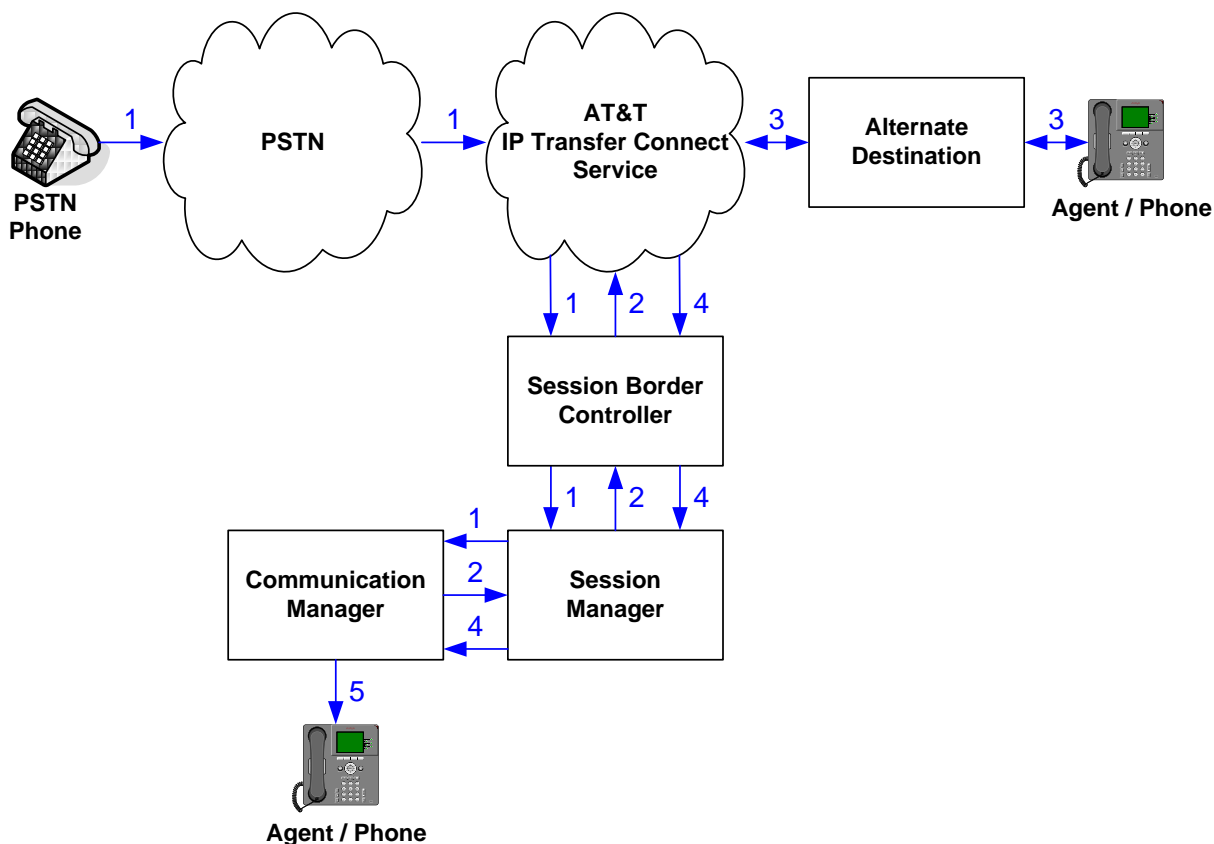
Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

3.2.4. Unsuccessful Inbound Refer Call to Alternate Destination

The third call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination. The fourth call scenario illustrated in the figure below is similar to the third call scenario, except that the redirection is unsuccessful due to the alternate destination being busy or otherwise unavailable. As a result, Communication Manager “takes the call back” and routes the call to an agent/phone.

1. Same as the third call scenario.
2. Same as the third call scenario.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The AT&T IP Transfer Connect service notifies the redirecting/referring party (Communication Manager) of the error condition.
5. Communication Manager routes the call to a local agent or phone.

Note: This “error handling” scenario occurs only with AT&T IP Transfer Connect service lines enabled with the Attended IP Courtesy Transfer feature.



Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Equipment/Software	Release/Version
Avaya S8800 Server	Avaya Aura® System Manager 6.2 SP2 (6.2.0.0.15669-6.2.12.202) System Platform 6.2.1.0.9
Avaya S8800 Server	Avaya Aura® Session Manager 6.2 SP2 (6.2.2.0.62205)
Avaya S8720 Server	Avaya Aura® Communication Manager 5.2.1 SP13 (02.1.016.4-19880)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW15 FW054
TN799DP Control-LAN (C-LAN)	HW01 FW040
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW062
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW011
Avaya 9650 IP Telephone	H.323 Version S3.11b
Avaya 9620C IP Telephone	H.323 version S3.11b
Avaya 9641G IP Telephone	H.323 Version S6.2013U
Avaya one-X® Agent	2.5.00467.18
Avaya Digital Telephone 6408D+	
Avaya Analog phone	-
Fax device	Ventafax Home Version 6.1.59.144
Acme Packet Net-Net 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Transfer Connect service using AVPN/MIS-PNT transport service connection	VNI 23

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release 6.2

The Session Manager administration for interoperability with AT&T IP Transfer Connect service is similar to the configuration done in [7] for AT&T IP Toll Free service. Refer to the appropriate section in [7] to configure the Session Manager for AT&T IP Transfer Connect service. Additional dial patterns and modification to the existing Routing Policy may be required for proper routing of the calls from AT&T Transfer Connect service.

6. Configure Avaya Aura® Communication Manager 5.2.1

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are similar to the AT&T IP Toll Free service configuration in [7]. This section describes additional administration step on Communication Manager to support interoperability with AT&T IP Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. For any values not configured, defaults are used in this reference configuration. These Application Notes assume that basic Communication Manager administration has already been performed. Consult [5] and [6] for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to this reference configuration. Other parameter values may or may not match based on local configurations.

6.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On **Page 4** of the **system-parameters customer-options** form, verify that the **ISDN/SIP Network Call Redirection** field is set to **y**.

display system-parameters customer-options		Page 4 of 11
OPTIONAL FEATURES		
Emergency Access to Attendant? y		IP Stations? y
Enable 'dadmin' Login? y		
Enhanced Conferencing? y		ISDN Feature Plus? y
Enhanced EC500? y	ISDN/SIP Network Call Redirection? y	
Enterprise Survivable Server? n		ISDN-BRI Trunks? y
Enterprise Wide Licensing? n		ISDN-PRI? y
ESS Administration? n	Local Survivable Processor? n	
Extended Cvg/Fwd Admin? y	Malicious Call Trace? n	
External Device Alarm Admin? n	Media Encryption Over IP? n	
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n	
Flexible Billing? n		
Forced Entry of Account Codes? n	Multifrequency Signaling? y	
Global Call Classification? n	Multimedia Call Handling (Basic)? y	
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y	
Hospitality (G3V3 Enhancements)? n	Multimedia IP SIP Trunking? n	
IP Trunks? y		
IP Attendant Consoles? n		

System-Parameters Customer-Options Form – Page 4

2. On **Page 6** of the **System-Parameters Customer-Options** form, verify that the vectoring features shown below are set to **y**.

display system-parameters customer-options	Page 6 of 11
CALL CENTER OPTIONAL FEATURES	
Call Center Release: 5.0	
ACD? y	Reason Codes? n
BCMS (Basic)? y	Service Level Maximizer? n
BCMS/VuStats Service Level? n	Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? n	Service Observing (Remote/By FAC)? n
Business Advocate? n	Service Observing (VDNs)? n
Call Work Codes? n	Timed ACW? n
DTMF Feedback Signals For VRU? n	Vectoring (Basic)? y
Dynamic Advocate? n	Vectoring (Prompting)? y
Expert Agent Selection (EAS)? y	Vectoring (G3V4 Enhanced)? y
EAS-PHD? y	Vectoring (3.0 Enhanced)? y
Forced ACD Calls? n	Vectoring (ANI/II-Digits Routing)? y
Least Occupied Agent? n	Vectoring (G3V4 Advanced Routing)? y
Lookahead Interflow (LAI)? n	Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n	Vectoring (Best Service Routing)? y
Multiple Call Handling (Forced)? n	Vectoring (Holidays)? n
PASTE (Display PBX Data on Phone)? n	Vectoring (Variables)? y

System-Parameters Customer-Options Form – Page 6

6.2. SIP Trunks

This section describes the additional steps for modifying SIP trunk to Session Manager to support AT&T IP Transfer Connect service.

1. Enter the **change trunk-group 2** command, where 2 is the trunk group configured in [7]. On **Page 4** of the **trunk-group** form, set the **Network Call Redirection** field to “y” (see **Item 2** in **Section 2.2**)

change trunk-group 2	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
Network Call Redirection? y	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 100	
Convert 180 to 183 for Early Media? y	
Always Use re-INVITE for Display Updates? n	
Enable Q-SIP? n	

Trunk Group form for IP Transfer Connect service calls – Page 4

6.3. Inbound Call Routing

This section describes the steps for routing inbound AT&T IP Transfer Connect service calls to reach Vector Directory Numbers with corresponding programmable vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality. The routing of inbound AT&T IP Toll Free service calls that do not invoke the SIP NCR functionality is addressed in [7].

Two different inbound call routing scenarios are described in these Application Notes:

- Pre-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP 302 message) prior to the call being answered.
- Post-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP REFER message) after the call has been answered by a vector.

These Application Notes provide rudimentary vector definitions simply necessary to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Application Enablement Services (AES) to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes. Consult [5] and [6] for further information.

6.3.1. Pre-Answer Redirection

This section provides an example of Pre-Answer Redirection. In this example, first the variables are defined for the vectors (**Figure 1**) in the inbound call is routed to the VDN shown in **Figure 2**, which invokes the vector shown in **Figure 3**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data “**1234567890123456**” to ASAI UUI variable “**A**” and “**7890123456789012**” to ASAI UUI variable “**B**” (vector steps **05** and **06**).
Note: The parameters for ASAI UUI variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 1**).
- Redirects the call to the number “**1092**” (vector step **08**). Note that since this vector did not answer the call, the presence of the “**~r**” in the “**route-to number**” instructs Communication Manager to send a SIP 302 message with the number “**1092**” in the user part of the Contact header URI (e.g., 1092@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered on the signaling group defined in [7] on which the inbound call arrives.

change variables							Page 1 of 39
VARIABLES FOR VECTORS							
Var	Description	Type	Scope	Length	Start	Assignment	VAC
A	UuiTest1	asaiuui	L	16	1		
B	UuiTest2	asaiuui	L	16	17		
C							

Figure 1: Change Variables Form

```

display vdn 6665316                                     Page 1 of 3
                VECTOR DIRECTORY NUMBER

                Extension: 6665316
                Name*: NCR Ringback 302 UUI
                Destination: Vector Number 16
                Attendant Vectoring? n
                Meet-me Conferencing? n
                Allow VDN Override? n
                COR: 1
                TN*: 1
                Measured: none

                VDN of Origin Annc. Extension*:
                1st Skill*:
                2nd Skill*:
                3rd Skill*:

* Follows VDN Override Rules

```

Figure 2: Sample VDN for Pre-Answer Redirection

```

display vector 16                                       Page 1 of 6
                CALL VECTOR

                Number: 16                               Name: NcrRedir_wUui
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? n      G3V4 Adv Route? y      CINFO? n      BSR? y      Holidays? n
  Variables? y      3.0 Enhanced? y
01 #      NCR Redirection with ringback and uui forwarding
02 wait-time 2 secs hearing ringback
03
04 #      Define UUI variable to send
05 set A = none CATR 1234567890123456
06 set B = none CATR 7890123456789012
07 #      Immediate redirect to AT&T speed dial number
08 route-to number ~r1092 with cov n if unconditionally
09
10 #      Play this announcement only on redirect failure
11 announcement 33008
12

```

Figure 3: Sample Vector for Pre-Answer Redirection

6.3.2. Post-Answer Redirection

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 4**, which invokes the vector shown in **Figure 5**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data “**1234567890123456**” to ASAI UI variable “**A**” and “**7890123456789012**” to ASAI variable “**B**” (vector steps **05** and **06**).
Note: The parameters for UI variable “**A**” and other vector variables are defined using the **change variables** command (see **Figure 1**).
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number “**1092**” (vector step **09**). Note that since this vector answered the call, the presence of the “**~r**” in the “**route-to number**” instructs Communication Manager to send a SIP REFER message with the number “**1092**” in the user part of the Refer-To header URI (e.g., 1092@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered on the signaling group defined in [7] on which the inbound call arrived.

```
display vdn 2018 Page 1 of 3
VECTOR DIRECTORY NUMBER
Extension: 2018
Name*: NCR Ringback REFER UII
Destination: Vector Number 18
Attendant Vectoring? n
Meet-me Conferencing? n
Allow VDN Override? n
COR: 1
TN*: 1
Measured: none

VDN of Origin Annc. Extension*:
1st Skill*:
2nd Skill*:
3rd Skill*:

* Follows VDN Override Rules
```

Figure 4: Sample VDN for Post-Answer Redirection

```
display vector 18                                     Page 1 of 6
                                                    CALL VECTOR

Number: 18                                           Name: NcrRefer_wUui
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
Prompting? y      LAI? n      G3V4 Adv Route? y      CINFO? n      BSR? y      Holidays? n
Variables? y      3.0 Enhanced? y
01 #      NCR Refer with ringback and uui forwarding
02 wait-time 2 secs hearing ringback
03
04 #      Define UUI variable to send
05 set A = none CATR 1234567890123456
06 set B = none CATR 7890123456789012
07 #      Refer to AT&T speed dial number
08 announcement 33007
09 route-to number ~r1092 with cov n if unconditionally
10 #      Play this announcement only on redirect failure
11 disconnect after announcement 33008
12
```

Figure 5: Sample Vector for Post-Answer Redirection

6.4. Saving Translations

To save all Communication Manager provisioning changes, enter the command **save translations**.

7. Configure Acme Packet Session Border Controller (SBC)

These Application Notes assume that basic Acme Packet SBC administration has already been performed. The Acme Packet SBC configuration used in the reference configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Use **putty** or similar tool to access Acme Packet SBC for configuration. Consult with Acme Packet Support [8] for further details and explanations on the configuration below.

ANNOTATION: The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Session Manager, Communication Manager, etc., reside to the AT&T IP services. The Session Agent Groups (**SAG**) defined here, and further down, provisioned under the session-groups **SP-PROXY** and **ENTERPRISE**. Note: Although **Enterprise** policy is not used for AT&T IP Transfer Connect service but is left in there in case the customer is using AT&T IP Flexible Reach service.

The Acme Packet SBC configuration for interaction with the AT&T IP Toll Free service is provided in [7]. The additional configuration on the Acme Packet SBC necessary for supporting interaction with the AT&T IP Transfer Connect service is provided below as a reference.

1. The AT&T IP Transfer Connect service requires that SIP Refer and 302 call redirection messages contain the AT&T Border Element IP address in the Refer-to header of a Refer call and the Contact header of a 302 call. The following section describes the Acme Packet header-rule provisioning. The header-rules described below were added to the existing sip-manipulation *NAT_IP* described in [7].

- **Modify Refer-to header**

```
header-rule
  name                modReferTo
  header-name         Refer-To
  action              manipulate
  comparison-type     case-sensitive
  msg-type            any
  methods             REFER
  match-value
  new-value
  element-rule
  name                modmline
  parameter-name
  type                uri-host
  action              find-replace-all
  match-val-type     any
  comparison-type     case-sensitive
  match-value         customerb.com
  new-value           $REMOTE_IP
```

- **Modify 302 Contact header**

```
header-rule
  name                mod302
  header-name         Contact
  action              manipulate
```

comparison-type	case-sensitive
msg-type	Reply
methods	INVITE
match-value	
new-value	
element-rule	
name	modmline
parameter-name	
type	uri-host
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	customerb.com
new-value	\$REMOTE_IP

- **Modify Sendonly** (See Item 2, Section 2.2 for further details)

header-rule		
name	modsendonly	
header-name	Content-type	
action	manipulate	
comparison-type	case-sensitive	
msg-type	any	
methods	INVITE	
match-value		
new-value		
element-rule		
name	modmline	
parameter-name	application/sdp	
type	mime	
action	find-replace-all	
match-val-type	any	
comparison-type	case-sensitive	
match-value	sendonly	
new-value	sendrecv	

The following shows the completed *NAT_IP* sip-manipulation with the additions described in item 1 above.

sip-manipulation	
name	NAT_IP
description	Topology hiding for TO and FROM headers
split-headers	
join-headers	
header-rule	
name	manipFrom
header-name	From
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	
match-value	
new-value	
element-rule	
name	FROM
parameter-name	
type	uri-host

action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	\$LOCAL_IP
header-rule	
name	manipTo
header-name	To
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	
match-value	
new-value	
element-rule	
name	TO
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	\$REMOTE_IP

header-rule	
name	modReferTo
header-name	Refer-To
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	REFER
match-value	
new-value	
element-rule	
name	modmline
parameter-name	
type	uri-host
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	customerb.com
new-value	\$REMOTE_IP

header-rule	
name	mod302
header-name	Contact
action	manipulate
comparison-type	case-sensitive
msg-type	Reply
methods	INVITE
match-value	
new-value	
element-rule	
name	modmline
parameter-name	
type	uri-host
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive

	match-value	customerb.com
	new-value	\$REMOTE_IP
header-rule		
	name	modsendonly
	header-name	Content-type
	action	manipulate
	comparison-type	case-sensitive
	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	element-rule	
	name	modmline
	parameter-name	application/sdp
	type	mime
	action	find-replace-all
	match-val-type	any
	comparison-type	case-sensitive
	match-value	sendonly
	new-value	sendrecv

8. Verification Steps

The following steps may be used to verify this reference configuration:

8.1. AT&T IP Transfer Connect

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Verify basic call functions such as hold, transfer and conference.
3. Verify the use of DTMF signaling.

8.2. Avaya Aura® Communication Manager

The following examples are only a few of the monitoring commands available on Communication Manager. See [5] and [6] for more information.

- From the Communication Manager console connection enter the command *list trace tac xxx*, where *xxx* is a trunk access code to verify that the inbound or outbound calls are using the right trunk groups. Similarly, *list trace station*, *list trace vdn*, and *list trace vector*, *status trunk* and *status station* commands can be used on Communication Manager.

8.3. Avaya Aura® Session Manager

Navigate to **Home**→ **Elements**→ **Session Manager**→ **System Status** → **SIP Entity Monitoring** and click on the SIP Entity for which the status is required. Following screen shows status for the entity link between Session Manager and Acme Packet SBC.

Note: The Reason Code column indicates that Session Manager has received a **SIP 405 Method Not Allowed** response (normal for this reference configuration) to the **SIP OPTIONS** it generated. This response is sufficient for SIP Link Monitoring to consider the link up.

The screenshot shows the Avaya Aura System Manager 6.2 web interface. The breadcrumb navigation is Home / Elements / Session Manager / System Status / SIP Entity Monitoring. The main content area is titled "SIP Entity, Entity Link Connection Status" and displays "All Entity Links to SIP Entity: AcmeSBCATT-5060". A table shows the connection status for one entity link:

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	DenverSM	10.80.130.250	5060	TCP	Up	405 Method Not Allowed	Up

9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager Network Call Redirection, and Acme Packet Session Border Controller can be configured to interoperate successfully with the AT&T IP Transfer Connect service. In addition, these Application Notes further demonstrate that the Avaya Aura® Communication Manager SIP Network Call Redirection (NCR) and User-to-User Information (UII) features can work in conjunction with the AT&T implementations of SIP NCR and UII to support call redirection over SIP trunks while preserving initiating caller information. This solution provides contact center users of Avaya Aura® Communication Manager the ability to redirect inbound AT&T IP Transfer Connect service calls to alternate destinations, and deliver UII-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide **configuration guidance** to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

10. References

The Avaya product documentation is available at <http://support.avaya.com> unless otherwise noted.

- [1] *Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.2, July 2012*
- [2] *Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.2, Issue 2, August 2012*
- [3] *Implementing Avaya Aura® System Manager, Doc ID 03-603473 Issue 1, July 2012*
- [4] *Administering Avaya Aura® System Manager, Document Number 03-603324, Release 6.2, July 2012*
- [5] *Administering Avaya Aura® Communication Manager, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509*
- [6] *Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Release 5.2, April 2009, Document Number 07-600780*
- [7] *Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session Manager 6.2 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0*

Acme Packet Support (login required):

- [8] <http://www.acmepacket.com/support.htm>

AT&T IP Transfer Connect Service Descriptions:

- [9] *AT&T IP Transfer Connect*
<http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/>

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